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IMPROVING THE QUALITY OF THE VOIP'S SERVICE

Abstract. *The paper is devoted to the introduction of the main parameters of Voice over Internet Protocol (VOIP) to improve the Quality of the Service (QoS). The main problem of VOIP is the quality of service. The article discusses various configurations of network quality settings to highlight parameter values for better VOIP performance. The quality of service is improved by selecting the correct priority types, voice codecs, and Differentiated Services Code Point (DSCP).*

Keywords: *VOIP, QoS, MOS, delay, jitter, DSCP, queuing methods.*

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УЛУЧШЕНИЕ КАЧЕСТВА ОБСЛУЖИВАНИЯ ГОЛОСОВОЙ СВЯЗИ ЧЕРЕЗ ИНТЕРНЕТ ПРОТОКОЛ

Аннотация. *Статья посвящена ознакомлению с основными параметрами голосовой связи через Интернет протокол для улучшения качества обслуживания. Главная проблема VOIP это качество обслуживания. В статье рассматриваются различные конфигурации настроек качества сети для*

выделения значений параметров для лучшей работы VOIP. Качество обслуживания улучшают выбор верных типов очередности, голосовых кодеков и значений полей кода дифференцирования траффика.

Ключевые слова: *Голосовая связь, качество обслуживания, средняя оценка качества речи, временные задержки, дрожание, поле кода дифференцирования траффика, формирование очереди.*

VoIP is the family of technologies that allows IP networks to be used for voice applications, such as telephony, voice instant messaging, and teleconferencing. VoIP and QoS have advanced extremely in the past several years. Research studies on Quality of Service in wireless, mobile, and IP networks have increased. The need for prioritizing voice traffic on IP networks has become imminent.

The purpose of the study is to systematize the possibilities of improving VOIP system to match PTSN in future.

QoS and security are the primary challenges of VoIP deployment. VoIP contains security issues for voice and video traffic which are normal in circuit switch organize, e.g. tapping and fake assaults and other IP related issues [1]. The quality of VoIP traffic can be dependent on the follow factors: queuing technics in traffic, Differentiated Services Code Point (DSCP), voice codecs and protocols. Furthermore, some researchers create their own VOIP architecture designs for high availability [2]. When the traffic transmitted over the network, some router implements queuing techniques define how packets are buffered. The experienced latency is also affected by queuing technique.

VoIP's poor QoS led researchers to examine the MOS, packet loss, jitter, latency and delay to improve VoIP's quality. These variables will be compared to reveal the best parameters in VOIP call customization. The voice call quality can be measured by Mean-Opinion Score which is based on a scale of 5(excellent) to 1(bad).

MOS for VoIP traffic using the main three queuing methods (First-IN-first-Out, Priority queuing, Weighted Fair Queuing) are demonstrated in Figure 1. The figure shows that the best queuing methods which give the best MOS are WFQ and PQ which is 4.3 [3].

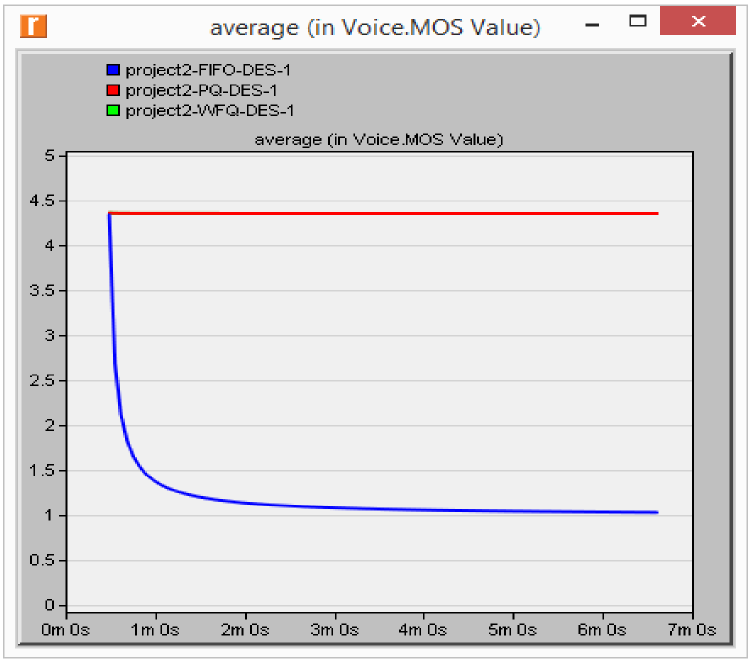

Figure 1 - Average MOS for queuing methods

Fig. 2 shws The VoIP packet End To End delay using the PQ and WFQ queue at the router interface is shown in Figure 2 [3]. ETE delays in PQ and WFQ in this experiment behave almost the same, there is no best queuing method. This value passes VoIP standards of the good ETE delay which is 150 ms.

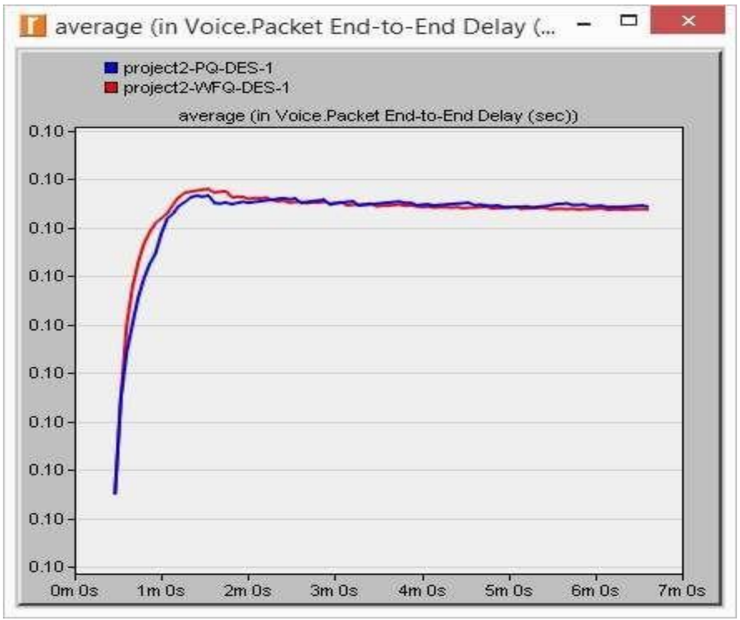

Figure 2 - End-to-End voice delay for WFQ and PQ

The traffic volume of transmitted and received File transfer protocol using WFQ and PQ methods are demonstrated in Figure 3 [3]. Average sent traffic in PQ and WFQ is the same, but received is higher in WFQ, this is because PQ gives absolute priority to real-time traffic compared FTP traffic.

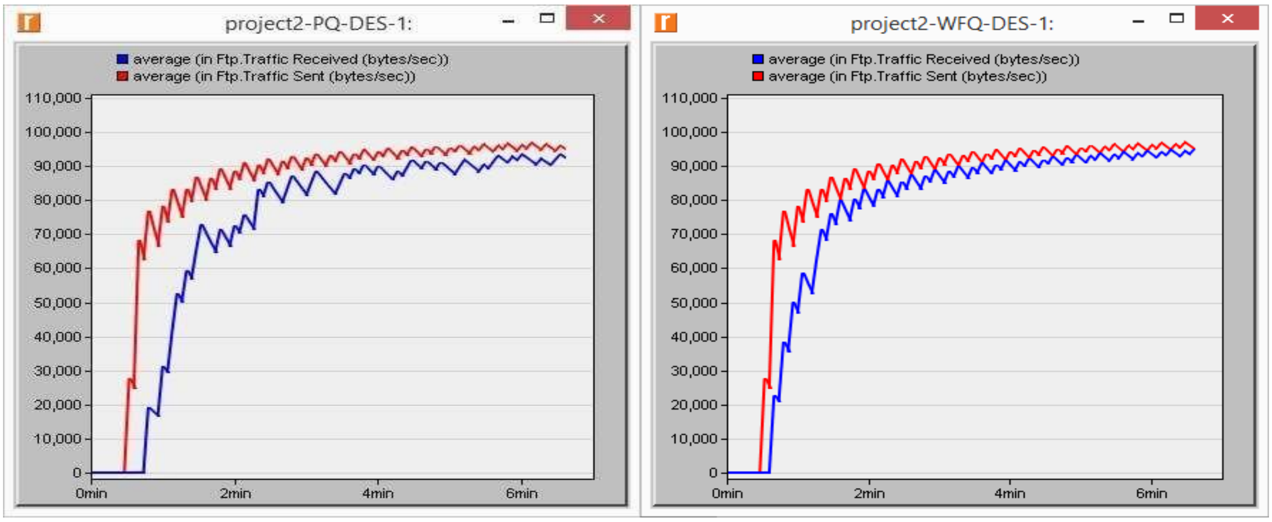

Figure 3 - FTP Transmitted/Received Traffic using PQ and WFQ

The VoIP MOS using three different codecs and WFQ method to declare the relation between using VoIP codec and voice quality are presented in Figure 4 [3]. The best voice codec which gives the best quality is G711, and this is logic because there is no voice compression. But G711 capacity is small because of its high consumed bandwidth. On the other hand, the figure shows that G729 gives better MOS compared to G723. Thus, the best codec from the view of capacity and quality is G729.

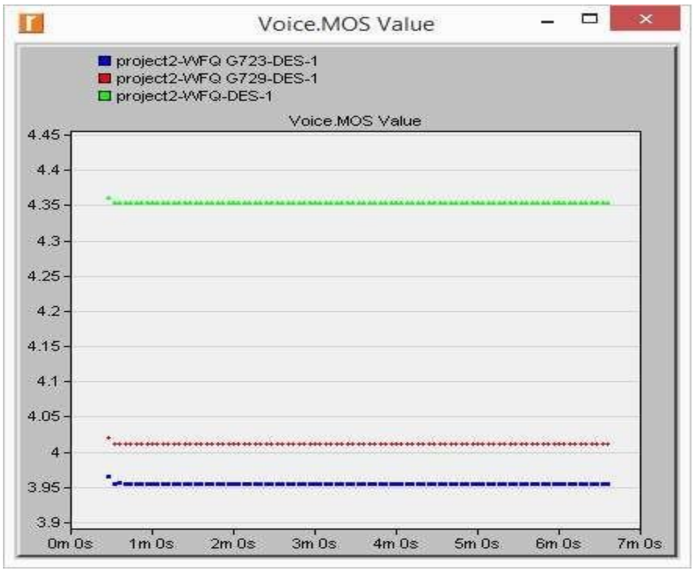

Figure 4 - Voice MOS for G711/G723/G729

Other research investigates the impact of changing DSCP markings on VoIP calls. The MOS' mean values for each of the test cases are illustrated in Figure 5. All the test cases have good call quality where none of the MOS values fell below the 4.3 scores [4]. The lowest mean value for MOS in this experiment is good quality VoIP phone conversation.

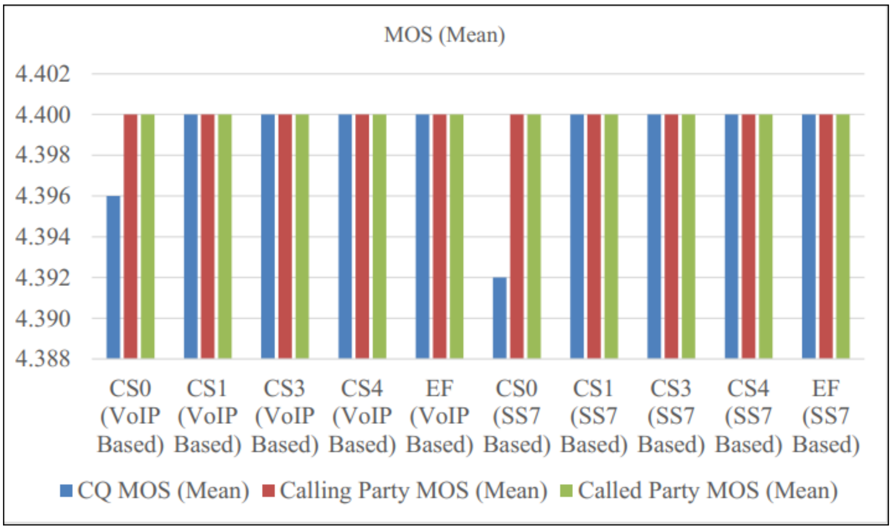

Figure 5 - MOS mean values for different DSCP marking

The average interpacket arrival time's mean values for each of the test cases are demonstrated in Figure 6. The mean value for the interpacket arrival time associated with each of the test cases is 20 ms [4]. These values passes VoIP standards of good interpacket arrival time which is 20 ms. The best results are given by CS3, CS4 and EF markings.

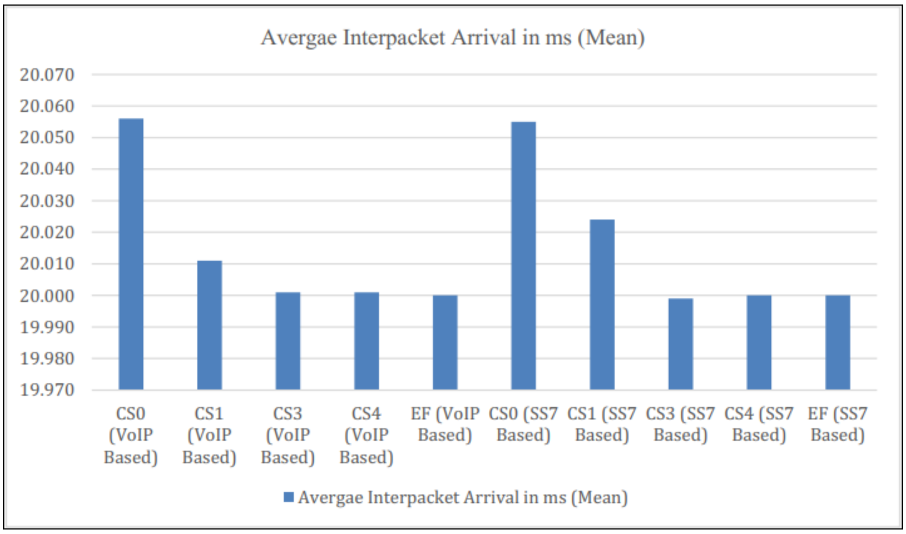

Figure 6 - Mean of the average interpacket arrival times in ms

The jitter's mean values for each of the test cases are illustrated in Figure 7. All the tested cases have good quality where none of the jitter values exceeded the 8 sec. The highest mean value for jitter is 3.77 ms that belongs to the CS1 marking of the VoIP-based manual test calls [4]. These values pass VoIP standards of the average jitter which is 50 ms. The higher VOIP traffic's DSCP marking value assignment is the lower jitter value is obtained. The best results are given by CS4 and EF markings.

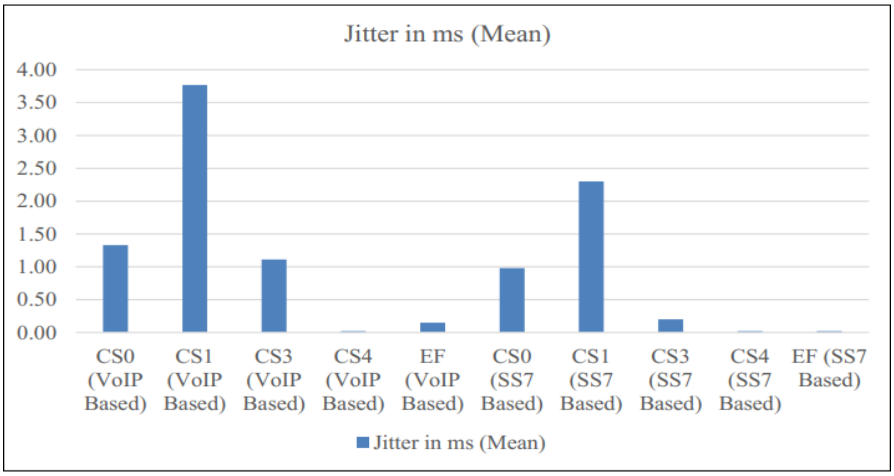

Figure 7 - Mean of the average jitter in ms

The mean values of latencies for each of the test cases are demonstrated in Figure 8. The highest mean value for the latency element is 23 ms that belongs to the CS0 marking of the VoIP-based manual test calls [4]. These values pass VoIP standards of latency which is 120 ms. The higher VOIP traffic's DSCP marking value assignment is, the lower latency value is obtained. The best results are given by CS4 and EF markings.

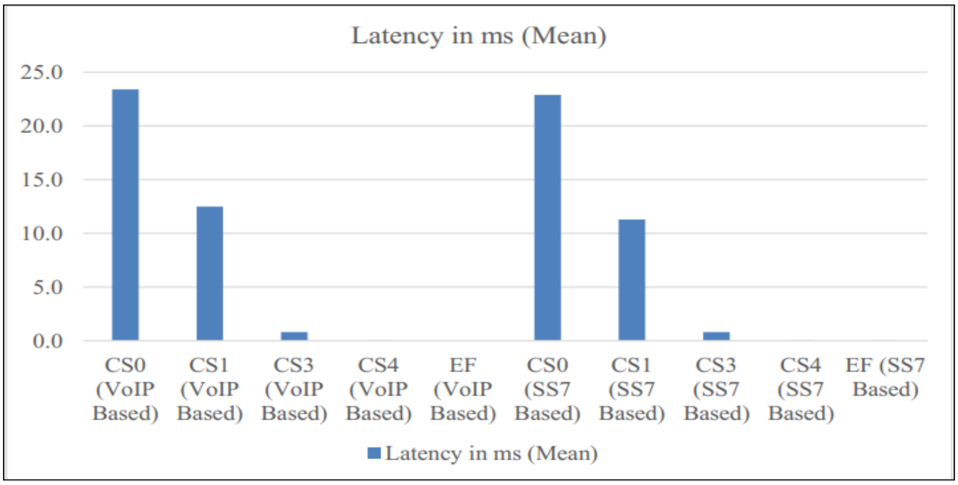

Figure 8 - Mean of the average latency in ms

Thus, configuring VoIP traffic with different queuing methods, voice codecs, DSCP marking actually improves VoIP's QoS. The best queuing method is WFQ because it gives minimum voice delay variation and ETE delay. Also, WFQ has an acceptable FTP throughput compared with PQ and FIFO. The study illustrates that the best codec from the view of capacity and quality is G729. This research demonstrates that the higher VoIP traffic's DSCP marking value assignment is, the less the chances of higher latency or packet loss in the VoIP traffic are possible. Configuring VoIP traffic with DSCP marking the EF and CS4 improve VoIP QoS.

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